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--2. (new) A fixed rate speech compression system for processing a frame of a speech signal, the fixed rate speech compression system comprising:

an encoder operable to encode a first part of the frame using common frame based encoding;

the common frame based encoding comprising pitch pre-processing to modify the waveform of the speech signal as a function of classification of the frame;

the encoder operable to select one of a first speech coding mode and a second speech coding mode to encode a second part of the frame.

3. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to continuously time warp the speech signal during pitch pre-processing when the frame is classified as indicative of increased voicing strength.

4. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to selectively perform continuous time warping of the speech signal during pitch pre-processing to introduce a variable delay of up to about twenty samples.

5. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to selectively estimate continuous time warping of the speech signal during pitch pre-processing by interpolation with Hamming weighted Sinc interpolation filters.

6. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to select the first speech coding mode as a function of classification of the frame as at least one of silence/background noise, noise-like unvoiced speech, unvoiced speech, onset speech, plosive speech and non-stationary voiced speech.

7. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to select the second speech coding mode as a function of classification of the frame as stationary voiced speech.

8. (new) The fixed rate speech compression system of claim 2, where a frame classified as at least one of background noise and unvoiced speech remains unchanged by pitch pre-processing.

9. (new) The fixed rate speech compression system of claim 2, where the encoder is operable to time shift the speech signal with pitch pre-processing in a frame classified as predominantly pulse-like unvoiced speech.

10. (new) A fixed rate speech compression system for processing a speech signal, the fixed rate speech compression system comprising:

an encoder operable to extract and encode parameters of a frame of the speech signal as a function of selection of one of a first mode and a second mode,

the encoder operable to jointly encode a pitch gain and a fixed codebook gain when the first mode is selected,

the encoder operable to delay separate encoding of the fixed codebook gain when the second mode is selected, the encoding of the fixed codebook gain delayed until the pitch gain is separately encoded and additional processing is performed by the encoder.

11. (new) The fixed rate speech compression system of claim 10, further comprising a two-dimensional vector quantization gain codebook operable with the encoder, where the encoder is operable to jointly encode the pitch gain and the fixed codebook gain with the two-dimensional vector quantization gain codebook when the first mode is selected.

12. (new) The fixed rate speech compression system of claim 10, further comprising a three-dimensional vector quantization block operable with the encoder, where the encoder is operable to separately encode the pitch gain with the three-dimensional vector quantization block when the second mode is selected.

13. (new) The fixed rate speech compression system of claim 10, further comprising a three-dimensional vector quantization gain codebook operable with the encoder, where the encoder is operable to separately encode the fixed codebook gain with the three-dimensional vector quantization gain codebook when the second mode is selected.

14. (new) The fixed rate speech compression system of claim 10, where the frame comprises a plurality of subframes, the encoder operable to jointly encode the pitch gain and the fixed codebook gain on a subframe basis.

15. (new) The fixed rate speech compression system of claim 10, where the encoder is operable to separately encode the pitch gain and the fixed codebook gain on a frame basis.

16. (new) The fixed rate speech compression system of claim 10, where the frame comprises a plurality of subframes, and the fixed codebook gain from each of the subframes are jointly encoded when the second mode is selected.

17. (new) The fixed rate speech compression system of claim 10, where the first speech coding mode comprises a first framing structure and the second speech coding mode comprises a second framing structure.

18. (new) A fixed rate speech compression system for processing a speech signal, the fixed rate speech compression system comprising:

an encoder operable to select one of a first mode and a second mode for a frame of the speech signal; and

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a fixed codebook comprising a plurality of sub codebooks operable with the encoder,

the encoder operable to select a final fixed codebook excitation represented by one of the sub codebooks, the final fixed codebook excitation selectable as a function of weighting applied to a best candidate fixed codebook excitation identified in each of the sub codebooks.

19. (new) The fixed rate speech compression system of claim 18, further comprising an adaptive first order filter operable with the encoder, the encoder operable to adapt perceptual weighting of the frame with the adaptive first order filter as a function of an instantaneous noise-to-signal ratio.

20. (new) The fixed rate speech compression system of claim 19, where the adaptive first order filter is modifiable by the encoder as a function of selection of one of the first mode and the second mode, modification of the adaptive first order filter operable to build characteristics into the frame prior to identification of the best candidate fixed codebook excitation in each of the sub codebooks.

21. (new) The fixed rate speech compression system of claim 19, where the adaptive first order filter is disabled when the frame of the speech signal is greater than about 2.5dB above a noise floor.

22. (new) The fixed rate speech compression system of claim 19, where the adaptive first order filter is operable to improve similarities between fixed codebook excitation represented in the sub codebooks and high frequency energy in high level background noise segments of the frame.

23. (new) The fixed rate speech compression system of claim 20, where the characteristics comprise a first characteristic and a second characteristic, the first characteristic introduced with a phase dispersion filter operable with the encoder when the first mode is selected, the second characteristic adaptively insertable by the encoder as a function of a correlation factor from a past frame when one of the first mode and the second mode is selected.

24. (new) The fixed rate speech compression system of claim 18, where the sub codebooks comprise a two pulse codebook, a three pulse codebook and a gaussian codebook when the first mode is selected.

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25. (new) The fixed rate speech compression system of claim 18, where the sub codebooks comprise a two pulse codebook, a three pulse codebook and a 6 pulse codebook when the second mode is selected.

26. (new) The fixed rate speech compression system of claim 18, where the encoder is operable to apply the weighting and compare the weighted mean square error of the best candidate fixed codebook excitation identified in each of the sub codebooks to select the final fixed codebook excitation.

27. (new) A fixed rate speech compression system for processing a speech signal, the fixed rate speech compression system comprising:

an encoder operable to select one of a first mode and a second mode to encode parameters of a frame of the speech signal and form a bitstream;

the parameters represented in a bit allocation comprising:

a mode representative of one of the first mode and the second mode; and

a pitch gain and a fixed codebook gain jointly encoded and represented jointly in the bit allocation when the first mode is selected, the fixed codebook gain and the pitch gain exclusively encoded and represented as separate bit allocations when the second mode is selected.

28. (new) The fixed rate speech compression system of claim 27, where the bit allocation further comprises a fixed codebook excitation represented by an entry in one of a two pulse codebook, a three pulse codebook and a gaussian codebook when the first mode is selected.

29. (new) The fixed rate speech compression system of claim 28, where the bit allocation of the fixed codebook excitation comprises 30 bits per frame when the first mode is selected.

30. (new) The fixed rate speech compression system of claim 27, where the bit allocation further comprises a fixed codebook excitation represented by an entry in one of a two pulse codebook, a three pulse codebook and a six pulse codebook when the second mode is selected.

31. (new) The fixed rate speech compression system of claim 30, where the bit allocation of the fixed codebook excitation comprises 39 bits per frame when the second mode is selected.

32. (new) The fixed rate speech compression system of claim 27, where the jointly encoded pitch gain and fixed codebook gain comprises 14 bits per frame when the first mode is selected.

33. (new) The fixed rate speech compression system of claim 27, where the exclusively encoded pitch gain comprises 4 bits per frame and the exclusively encoded fixed codebook gain comprises 8 bits per frame when the second mode is selected.

34. (new) A method of processing a frame of a speech signal with a fixed rate speech compression system, the method comprising:

encoding a first part of the frame with common frame based encoding, the common frame based encoding comprising:

classifying the frame;

pitch pre-processing to modify the waveform of the speech signal as a function of classification of the frame; and

selecting one of a first speech coding mode and a second speech coding mode to encode a second part of the frame.

35. (new) The method of claim 34, where classifying the frame comprises classifying the frame as a function of pitch correlation information.

36. (new) The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as indicative of increased voicing strength; and

continuously time warping the frame of the speech signal to introduce a variable delay.

37. (new) The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as predominantly pulse-like unvoiced speech; and

time shifting the waveform as a function of an accumulated delay.

38. (new) The method of claim 34, where pitch pre-processing comprises:

classifying the speech signal as at least one of predominantly background noise and predominantly unvoiced speech; and

resetting an accumulated delay without modification of the waveform.

39. (new) The method of claim 34, where pitch pre-processing comprises modifying at least one pitch cycle of the speech signal to provide continuous time warping of the speech signal.

40. (new) The method of claim 34, where selecting the first speech coding mode comprises classifying the frame as at least one of silence/background noise, noise-like unvoiced speech, unvoiced speech, onset speech, plosive speech and non-stationary voice speech.

41. (new) The method of claim 34, where selecting the second speech coding mode comprises classifying the frame as stationary voiced speech.

42. (new) A method of processing a speech signal with a fixed rate speech compression system, the method comprising:

extracting parameters of a frame of the speech signal;

encoding a portion of the parameters with common frame based coding;

selecting one of a first mode and a second mode;

jointly encoding a pitch gain and a fixed codebook gain when the selection is the first mode; and

encoding the pitch gain separate from the fixed codebook gain when the selection is the second mode.

43. (new) The method of claim 42, where jointly encoding the pitch gain and the fixed codebook gain comprises identifying an entry in a two-dimensional vector quantization gain

codebook, the entry representative of the pitch gain and a correction factor for a predicted fixed codebook gain.

44. (new) The method of claim 42, where the predicted fixed codebook gain is determined with prediction coefficients and moving average prediction of a fixed codebook energy.

45. (new) The method of claim 42, where encoding the pitch gain separate from the fixed codebook gain comprises:

quantizing the pitch gain; and

subsequently encoding the fixed codebook gain.

46. (new) The method of claim 42, where encoding the pitch gain separate from the fixed codebook gain comprises deriving the pitch gain with open loop quantization during pitch pre-processing.

47. (new) A method of processing a speech signal with a speech compression system, the method comprising:

selecting one of a first framing structure and a second framing structure to encode parameters of a frame of the speech signal;

predicting a fixed codebook characteristic for each of a plurality of subframes when the second framing structure is selected, the prediction performed as a function of prediction coefficients associated with each subframe and a fixed codebook characteristic from each of a plurality of subframes of a previous frame; and

representing a fixed codebook gain for each of the subframes with the respective predicted fixed codebook characteristic when the second framing structure is selected.

48. (new) The method of claim 47, further comprising:

deriving a pitch gain for each of a plurality of subframes during pitch pre-processing;

quantizing the pitch gain of each of the subframes;

storing the quantized pitch gain of each of the subframes; and

performing delayed joint quantization of the fixed codebook gains for each of the subframes as a function of the stored quantized pitch gain of each of the subframes.

49. (new) The method of claim 47, where predicting the fixed codebook characteristic comprises applying third order moving average prediction.

50. (new) The method of claim 47, where the prediction coefficients comprise a first subframe predictor coefficient represented as {0.6, 0.3, 0.1}, a second subframe predictor coefficient represented as {0.4, 0.25, 0.1} and a third subframe predictor coefficient represented as {0.3, 0.15, 0.075}.

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51. (new) The method of claim 47, where the fixed codebook characteristic comprises fixed codebook energy.--